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[10191/1822]

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
BOARD OF PATENT APPEALS AND INTERFERENCES

In re Application of:	: Examiner: Michael N. Opsasnick
Torsten Prange et al.	
For: METHOD FOR CODING OR DECODING SPEECH SIGNAL SAMPLED VALUES, AND CODER OR DECODER	
Filed: November 6, 2001	Art Unit: 2653
Serial No.: 09/807,015	

I hereby certify that this correspondence is being deposited with the United States Postal Service with sufficient postage as first class mail in an envelope addressed to: Mail Stop APPEAL BRIEF - PATENTS Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450 on

Date: 4/17/2007

Signature: AARON C. DEDITCH (33,865)

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P.O. Box 1450
Alexandria, VA 22313-1450

APPEAL BRIEF TRANSMITTAL AND PETITION TO EXTEND

SIR:

Accompanying this Appeal Brief Transmittal is an Appeal Brief pursuant to 37 C.F.R. § 41.37 **in triplicate** as a courtesy (even though not required) for filing in the above-identified patent application.

This is also a **Petition To Extend** Under 37 C.F.R. § 1.136(a) to extend the two-month response date by **(3) months** from the two-month date of January 17, 2007 to April 17, 2007 (Appellants mailed a Notice Of Appeal on November 13, 2006 which was filed in the U.S.P.T.O. on November 17, 2006 so that the two-month appeal brief due date is January 17, 2007).

Please charge the appropriate fees of **$1,520.00**, which includes the Appeal Brief fee under 37 C.F.R. § 1.17(c) (which is believed to be $500.00) and the Rule 136(a) extension fee (which is believed to be $1,020.00 for a three-month extension), to Deposit Account No. **11-0600**. The Commissioner is also authorized, as necessary and/or appropriate, to charge any additional and appropriate fees, including any further Rule 136(a) extension fees, or credit any overpayment to Deposit Account No. **11-0600**. Two duplicate copies of this transmittal are enclosed for these purposes.

Dated: 4/17/2007

Respectfully submitted,

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It is understood for purposes of the appeal that any Amendments to date have already been entered by the Examiner, and that the Response After Final does not require entry since it included no amendments.

As to the length of the "concise explanation" of the subject matter defined in each of the claims involved in the appeal (see 41.37), the "concise explanation" language is like the "concise explanation" requirement of former Rule 37 C.F.R. § 1.192. Accordingly, the length of the concise explanation provided is acceptable, since it would have been acceptable under 37 C.F.R. § 1.192 and since it specifically defines the subject matter of the independent claims involved and in the appeal. In the filing of many appeal briefs under the old rule for the present Assignee, the length of the "concise explanation" has always been ultimately accepted by the Patent Office.

It is therefore respectfully submitted that this Appeal Brief complies with 37 C.F.R. § 41.37. Although no longer required by the rules, this Brief is submitted in triplicate as a courtesy to the Appeals Board.

It is respectfully submitted that the final rejections of claims 11 to 19 and 21 to 24 should be reversed for the reasons set forth below.

1. REAL PARTY IN INTEREST

The real party in interest in the present appeal is Robert Bosch GmbH ("Robert Bosch") of Stuttgart in the Federal Republic of Germany. Robert Bosch is the assignee of the entire right, title and interest in the present application.

2. RELATED APPEALS AND INTERFERENCES

There are no interferences or other appeals related to the present application, which "will directly affect or be directly affected by or have a bearing on the Board's decision in the pending appeal".

3. STATUS OF CLAIMS

CLAIMS 1 TO 10 and 20 ARE CANCELED.

A. Claims 11 to 19 and 22 to 24 were rejected under 35 U.S.C. § 102(e) as anticipated by Fette et al., U.S. Patent No. 5,797,121.

B. Claim 20 was rejected under 35 U.S.C. § 103(a) as unpatentable over "Fette" in view of Gersho et al, U.S. Patent No. 6,233,550.

C. Claim 21 was rejected under 35 U.S.C. § 103(a) as unpatentable over "Fette" in view of Tzeng, U.S. Patent No. 5,797,121.

Appellants therefore appeal from the final rejections of pending and considered claims 11 to 19 and 21 to 24. A copy of all of the pending and considered and appealed claims 11 to 19 and 21 to 24 is attached hereto in the Claims Appendix.

4. STATUS OF AMENDMENTS

In response to the Final Office Action mailed on May 11, 2006, Appellants filed a Response After A Final Office Action (with no amendments), which was mailed on November 13, 2006.

It is understood for purposes of the appeal that any Amendments to date have already been entered by the Examiner, and that the Response After Final does not require entry since it included no amendments.

5. SUMMARY OF CLAIMED SUBJECT MATTER

The concise explanation of the summary of the claimed subject matter is as follows, as described in the context of the present application.

As in claims 11, 21 and 22, the specification and Figures disclose and describe the following:

In the HVXC speech decoder of Figure 1, the transmitted speech LPC parameters, the voiced/unvoiced decision of the encoder, and the excitation parameters are read out from the bitstream and are supplied to inputs 1, 2, and 3. The LPC parameters contain indices from which inverse LSP vector quantizer 16 regenerates the LSP (Line Spectral Pairs) parameters. For this purpose, LSP code books 4 (CbLsp) and 5 (CbLsp4) are indexed with the LPC parameters, and the LSP parameters are read out. Dependent on the voiced/unvoiced decision of this frame, if necessary, interpolation – module 6 – takes place between the LSP parameters of the past and current frame. Conversion then occurs into LPC parameters, which enter as coefficients into the LPC synthesis filter – modules 7 and 8. (See specification, page 2, line 26 to page 3, line 4).

Parallel to this, and as a function of the voiced/unvoiced decision, the vectors for the spectral envelope (voiced frame), AM code books 9 (CbAm) and 10 (CbAm4), or the vectors for the stochastic excitation signal (unvoiced frame, CELP code books 11 (CbCelp) and 12 (CbCelp4)) are read. The regeneration occurs using the inverse vector quantizers 13 and 14. After the harmonic synthesis (voiced) – module 15 – the filtering of the speech data takes place in the LPC synthesis filter. The output data from the voiced - module 7 - and from the unvoiced - module 8 - synthesis filter are subsequently added to get the reconstructed speech signal. (See specification, page 3, lines 6 to 13).

Because values for the code books in floating-point form are not suitable for fixed-point DSPs, and because the required word lengths would be too large, conversion of the table values for the code books that were obtained from the speech signal sampled values takes place in a quantized form, with resulting equivalent speech quality. The word lengths required for this for the individual table values are determined in various hearing tests. The quantization is to a *wordlength*, and is expressed in bits. A signed whole number having *wordlength* bits includes a value range from $-2^{wordlength-1}$ to $2^{wordlength-1}-1$. The quantization of the code books is done as shown below. For "Study on ISO/IEC 14496-3 FCD, Subpart 3", the code book *cb* is defined as follows: $cb = \{a0, a1, \ldots, an, \ldots, am\}$ with $0 \leq n \leq m$ and $an0R$. (See specification, page 3, lines 15 to 30).

For the quantization of the individual elements, the following is done:

As to the determination of the value range of the code books of the independent claims, to obtain a well-matched quantization, the elements of each code book are scaled so that the available value range is exploited, where the value range of the elements is between:

$$\frac{-2^{wordlength-1}}{2^{wordlength-1}} = -1 \quad \textbf{and} \quad \frac{2^{wordlength-1}-1}{2^{wordlength-1}} = 1 - 2^{-(wordlength-1)}$$

To achieve this, the maximum of the positive and of the negative elements (*max_pos* or *max_neg*) of each code book is determined, and are from:

$max\_pos = \max\left(\{a_n \in cb \mid a_n \geq 0\}\right)$ or $max\_neg = \min\left(\{a_n \in cb \mid a_n \geq 0\}\right)$, with $0 \leq n \leq m$. (See specification, page 4, lines 1 to 12).

As to the multiplication feature of claims 11, 21 and 22, as *a function of max_pos or max_neg, the following steps result: max_pos > $(1 - 2^{-(wordlength-1)})$ or max_neg < -1, where max_pos and max_neg are multiplied by ½. If the result still satisfies the condition set under (a), then the process is repeated until the condition no longer holds. The number of multiplications by ½ is counted and is stored in the variables scale, so that: max_pos $\leq(1 - 2^{-(wordlength-1)})$ or max_neg $\geq$-1, where max_pos and max_neg are multiplied by 2. If the result still satisfies the condition set under (b), then the process is repeated until the condition no*

longer holds. The number of multiplications by 2 is counted and is stored in the variables scale. (See specification, page 4, lines 13 to 22).

2. Scaling of the elements of *cb* to the range between -1 and $(1 - 2^{-(wordlength - 1)})$. As a function of the decision of 1, the scaling of all code book entries is to the cited range:

$$b_n = \frac{1}{2^{scale}} a_n \forall a_n \in cb \quad \text{with } 0 \leq n \leq n$$

$$b_n = 2^{scale} a_n \forall a_n \in cb \quad \text{with } 0 \leq n \leq n.$$

After, the entries of each code book are in the following range: $-1 \leq b_n \leq (1 - 2^{-(wordlength - 1)})$, with $0 \leq n \leq n$. (See specification, page 4, line 24 to page 5, line 10).

3. *As to the scaling to wordlength bits of the independent claims*, for the scaling to the required value range, multiplication by $2^{wordlength - 1}$ takes place. In this way, the values of code books c^n are located in the range between $-2^{wordlength - 1}$ and $2^{wordlength - 1} - 1$. (See specification, page 5, lines 7 to 10).

4. Rounding - Before the decimal places are truncated, the determined entries are rounded so that depending on the sign + 0.5 or – 0.5 is added, so that: $c_n \geq 0 : d_n = c_n + 0.5$ and $c_n < 0 : d_n = c_n - 0.5$. Care is to be taken not to exceed the maximum permissible value range. This is located in the range as indicated under 2. (See specification, page 5, lines 12 to 20).

5. *As to indistinguishable speech quality*, the final quantization is through the truncation of the decimal places, so that with the variables *wordlength* at 16, *a speech quality indistinguishable from the original is obtained*. (See specification, page 5, lines 22 to 27).

In Figure 2, the block switching diagram of a CELP decoder is shown. First, the elements for decoding a frame are read from a transmitted bitstream. The parameters (elements) are supplied to decoder inputs 17 to 21. The excitation parameters are made up of the parameters for adaptive code book (lag) 22 for the generation of periodic signal components (voiced) and the parameters for fixed code books (shape index) 23a . . . 23n. (See specification, page 6, lines 1 to 7).

As to the multiplication feature of the independent claims, the entries of fixed code books 23a ... 23n and of adaptive code book 22 are each multiplied by a scaling factor (gain) via gain decoder 24. This scaling factor is reconstructed with the gain indices present at the input 21 and the gain VQ (vector quantization) tables stored in code books 25. The finally valid excitation vector is from the sum of the fixed and the adaptive code book vector. (See specification, page 6, lines 9 to 14).

With the vector quantizer VQ, the LPC indices represent the vector-quantized LSP (Line Spectral Pairs) parameters. The vectors of the first and second stage of the inverse vector quantization of the LSP parameters are obtained by reading out the LSP-VQ table values, which are stored in code books 26. Inside each frame, for each subframe interpolation - module 28 - occurs between the LSP parameters of the past and of the current frame. The LSP parameters, converted into LPC parameters, enter into LPC synthesis filter 29 as coefficients. The reconstruction of the speech data takes place there through filtering of the excitation signal. (See specification, page 6, lines 15 to 24).

The LSP VQ table values, and the gain VQ table values for code books 25 and 26 are normally present in a floating-point representation, which is not suitable for fixed-point DSP processing. For the same reasons as with the HVXC decoder (Figure 1), a conversion of the table values into a quantized form takes place, as before. (See specification, page 6, lines 26 to 32).

In summary, the presently claimed subject matter is to a method for one of coding and decoding speech signal sampled values, including: quantizing values previously obtained by an analysis from the speech signal sampled values and used for a generation of speech signal parameters before being stored in code books/code tables, the quantizing occurring to a word length that results in no noticeable losses in speech quality; storing in the code books/code tables the values previously obtained by the analysis from the speech signal sampled values and used for the generation of speech signal parameters; scaling the values of each code book/code table such that an available range of values is exploited as completely as possible, the scaling including: determining a maximum of a positive value and a negative value of each code book/code table, if the available range of values is exceeded,

performing a multiplication of the values of each code book/code table by a first factor smaller than one, and repeating the multiplication until all elements are located in the available range of values; and causing a number of repeated multiplications to be used as a scaling factor for all code book/code table entries, in which for a HXVC (Harmonic Vector Excitation Coding) speech coder/speech decoder, LPC coefficients, spectral envelopes of a speech signal, and unvoiced segments of the speech signal are stored in quantized form in corresponding ones of the code books/tables. (See claim 11).

The presently claimed subject matter is also to a method for one of coding and decoding speech signal sampled values, including: quantizing values previously obtained by an analysis from the speech signal sampled values and used for a generation of speech signal parameters before being stored in code books/code tables, the quantizing occurring to a word length that results in no noticeable losses in speech quality; storing in the code books/code tables the values previously obtained by the analysis from the speech signal sampled values and used for the generation of speech signal parameters; scaling the values of each code book/code table such that an available range of values is exploited as completely as possible, the scaling including: determining a maximum of a positive value and a negative value of each code book/code table, if the available range of values is exceeded, performing a multiplication of the values of each code book/code table by a first factor smaller than one, and repeating the multiplication until all elements are located in the available range of values; and causing a number of repeated multiplications to be used as a scaling factor for all code book/code table entries, in which: for a CELP (Code Excited Linear Prediction) speech coder/decoder, values for LSP (Line Spectral Pairs) VQ vector quantization code book/table entries, as well as those of gain VQ table entries, are stored in quantized form. (See claim 21).

*The presently claimed subject matter is also to a*n apparatus corresponding to one of a coder and a decoder for processing speech signal sampled values in accordance with a method of analysis through synthesis, comprising: an arrangement for storing in quantized form values contained in code books/code tables for a generation of speech signal

parameters; an arrangement for selecting a word length such that no noticeable losses in speech quality occur; an arrangement for quantizing the values contained in the code books/code tables to the word length that results in no noticeable losses in speech quality; an arrangement for scaling the values of each code book/code table such that an available range of values can be exploited as completely as possible; an arrangement for determining a maximum of positive values and negative values of each code book/code table, and for multiplying the values of each code book/code table by a first factor less than one if the available range of values is exceeded; and an arrangement for, if a multiplication of the values of the code books/code tables lies outside the available range of values, performing a repeated multiplication until all elements are located in the available range of values, and for providing a number of repeated multiplications as a scaling factor. *(See claim 22).*

Finally, the appealed claims include no means-plus-function language and no step-plus-function claims, so that 37 C.F.R. 41.37(v) is satisfied as to its specific requirements for such claims, since none are present here. Also, the present application does not contain any step-plus-function claims because the method claims in the present application are not "step plus function" claims because they do not recite "a step for", as required by the Federal Circuit and as stated in Section 2181 of the MPEP.

6. GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL

A. Whether claims 11 to 19 and 22 to 24 are anticipated under 35 U.S.C. § 102(e) by Fette et al., U.S. Patent No. 5,797,121.

B. Whether claim 20 is unpatentable under 35 U.S.C. § 103(a) over "Fette" in view of Gersho et al, U.S. Patent No. 6,233,550.

C. Whether claim 21 is unpatentable under 35 U.S.C. § 103(a) over "Fette" in view of Tzeng, U.S. Patent No. 5,797,121.

7. ARGUMENT

A. The Rejections Under 35 U.S.C. § 102(e) That Claims 11 to 19 and 22 to 24 Are Anticipated By "Fette"

Claims 11 to 19 and 22 to 24

Claims 11 to 19 and 22 to 24 were rejected under 35 U.S.C. § 102(e) as anticipated by Fette et al., U.S. Patent No. 5,797,121.

As regards the anticipation rejections of the claims, to reject a claim under 35 U.S.C. § 102(b), the Office must demonstrate that each and every claim feature is identically described or contained in a single prior art reference. (See *Scripps Clinic & Research Foundation v. Genentech, Inc.*, 18 U.S.P.Q.2d 1001, 1010 (Fed. Cir. 1991)). As explained herein, it is respectfully submitted that the Office Actions to date do not meet this standard, for example, as to all of the features of the claims. Still further, not only must each of the claim features be identically described, an anticipatory reference must also enable a person having ordinary skill in the art to practice the claimed invention, namely the claimed subject matter of the claims, as discussed herein. (See *Akzo, N.V. v. U.S.I.T.C.*, 1 U.S.P.Q.2d 1241, 1245 (Fed. Cir. 1986)).

As further regards the anticipation rejections, to the extent that the Office Actions to date may be relying on the inherency doctrine, it is respectfully submitted that to rely on inherency, the Examiner must provide a "basis in fact and/or technical reasoning to reasonably support the determination that the allegedly inherent characteristics *necessarily* flows from the teachings of the applied art." (See M.P.E.P. § 2112; emphasis in original; and see *Ex parte Levy*, 17 U.S.P.Q.2d 1461, 1464 (Bd. Pat. App. & Int'f. 1990)). Thus, the M.P.E.P. and the case law make clear that simply because a certain result or characteristic may occur in the prior art does not establish the inherency of that result or characteristic. Accordingly, it is respectfully submitted that any anticipation rejection premised on the inherency doctrine is not sustainable absent the foregoing conditions.

The "Fette" reference refers to performing scalar quantization in dividing the minimum value to maximum value ranges for each speech parameter into 2^N discrete

intervals. So long as speech quality degradation is not significant, there is a reduction of value N by 1 so as to quantize with one less bit of precision than in the previous iteration. As to this description, it does not identically describe (or even suggest in the context of claim 11 or claims 20 and 21) the feature of claim 11, which provides that: "if the available range of values is exceeded, *performing a multiplication of the values of each code book/code table by a first factor smaller than one, and repeating the multiplication until all elements are located in the available range of values; and causing a number of repeated multiplications to be used as a scaling factor for all code book/code table entries*. The Final Office Action has not shown because it cannot show that "Fette" uses the number of repeated multiplications in an iteration process as a scaling factor, as provided for in the context of claim 11 (and claims 20 and 21).

Furthermore, the "Fette" reference (as well as Gersho and Tzeng) does not identically describe (or even suggest) the feature of storing speech signals in quantized form in code books/tables, in which the speech signals are HXVC, LPC coefficients, spectral envelopes and unvoiced segments with such a scaling factor. (Gersho only refers to a harmonic coder where LP residuals and spectral magnitudes are quantized, but is completely silent about using a scaling factor as provided for in the context of the claimed subject matter of claims 11, 20 and 21).

As to the "multiplication" obtained "scaling factor", it is axiomatic that the terms of a claim are not interpreted in a vacuum, even though a pending claim may be "given the broadest reasonable interpretation *consistent with the specification*." M.P.E.P. § 2111. The law supports the eminently reasonable interpretation of the terms discussed herein based on the specification, as explained above. (See *In re Weiss*, 26 U.S.P.Q.2d 1885, 1887 (Fed. Cir. 1993) (when interpreting a claim term or phrase, one must "look to the specification for the meaning ascribed to that term"; Board reversed) (unpublished decision); *In re Okuzawa*, 190 U.S.P.Q. 464, 466 (C.C.P.A. 1976) ("claims are not to be read in a vacuum, and limitations therein are to be interpreted in light of the specification in giving them their broadest *reasonable* interpretation"; Board reversed; emphasis in original) (citing *In re Royka*, 180 U.S.P.Q. 580, 582-83 (C.C.P.A. 1974) (claims are "not to be read in a vacuum and while it is

true that they are to be given the broadest reasonable interpretation during prosecution, their terms still have to be given the meaning called for by the specification of which they form a part"; Board reversed; emphasis in original); and *In re Rohrbacher*, 128 U.S.P.Q. 117, 119 (C.C.P.A. 1960) (an "applicant is his own lexicographer and words used in his claims are to be interpreted in the sense in which they are used in the specification"; Board reversed)). This applies to the "multiplication" obtained "scaling factor", as provided for in the context of the claimed subject matter and as disclosed as to the only embodiments of the specification and drawing, as explained herein and in the present application.

Accordingly, the "Fettte" reference does not identically disclose (or even suggest) all of the features of claim 11 – as it must to support an anticipation rejection, and therefore claim 11 is allowable.

Claim 12 to 19 depend from claim 11, and are therefore allowable for at least the same reasons as claim 11.

Withdrawal of the anticipation rejections is therefore respectfully requested.

B. The Rejection Under 35 U.S.C. § 103(a) That Claim 20 Is Unpatentable Over "Fette" In View Of "Gersho"

Claim 20

Claim 20 was rejected under 35 U.S.C. § 103(a) as unpatentable over "Fette" in view of Gersho et al, U.S. Patent No. 6,233,550.

Claim 20 has previously been canceled without prejudice, and its obviousness rejection is therefore moot. Accordingly, the claim 20 rejection should be withdrawn.

C. The Rejection Under 35 U.S.C. § 103(a) That Claim 21 Is Unpatentable Over "Fette" In View Of "Tzeng"

Claim 21 was rejected under 35 U.S.C. § 103(a) as unpatentable over "Fette" in view of Tzeng, U.S. Patent No. 5,797,121.

To reject a claim as obvious under 35 U.S.C. § 103, the prior art must disclose or suggest each claim feature and it must also provide a motivation or suggestion for combining

the features in the manner contemplated by the claim. (See Northern Telecom, Inc. v. Datapoint Corp., 908 F.2d 931, 934 (Fed. Cir. 1990), cert. denied, 111 S. Ct. 296 (1990); In re Bond, 910 F.2d 831, 834 (Fed. Cir. 1990)). Thus, the "problem confronted by the inventor must be considered in determining whether it would have been obvious to combine the references in order to solve the problem", Diversitech Corp. v. Century Steps, Inc., 850 F.2d 675, 679 (Fed. Cir. 1998).

In particular, in rejecting a claim under 35 U.S.C. § 103(a), the Examiner bears the initial burden of presenting a *prima facie* case of obviousness. In re Rijckaert, 9 F.3d 1531, 1532, 28 U.S.P.Q.2d 1955, 1956 (Fed. Cir. 1993). To establish prima facie obviousness, three criteria must be satisfied. First, there must be some suggestion or motivation to modify or combine reference teachings. In re Fine, 837 F.2d 1071, 5 U.S.P.Q.2d 1596 (Fed. Cir. 1988). This teaching or suggestion to make the claimed combination must be found in the prior art and not based on the application disclosure. In re Vaeck, 947 F.2d 488, 20 U.S.P.Q.2d 1438 (Fed. Cir. 1991). Second, there must be a reasonable expectation of success. In re Merck & Co., Inc., 800 F.2d 1091, 231 U.S.P.Q. 375 (Fed. Cir. 1986). Third, the prior art reference(s) must teach or suggest all of the claim features. In re Royka, 490 F.2d 981, 180 U.S.P.Q. 580 (C.C.P.A. 1974).

Claim 21

Claim 21 includes the "multiplication" obtained "scaling factor" like that of claims 11 and 21, and is therefore allowable for essentially the same reasons as claims 11 and 21. Also, it is respectfully submitted that even if it were proper to modify the primary reference as conclusorily suggested by the Final Office Action (which is not conceded), the secondary reference does not cure – and is not asserted to cure – the critical deficiencies of the primary "Fette" reference. Accordingly, claim 21 is allowable.

Claims 11 to 19 and 21 to 24 are therefore allowable.

CONCLUSION

In view of the above, it is respectfully requested that the rejections of the finally rejected claims 11 to 19 and 21 to 24 be reversed, and that these claims be allowed as presented.

Respectfully submitted,

Dated: 4/17/2007

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CLAIMS APPENDIX

1-10. (Canceled).

11. (Previously Presented) A method for one of coding and decoding speech signal sampled values, comprising the steps of:

quantizing values previously obtained by an analysis from the speech signal sampled values and used for a generation of speech signal parameters before being stored in code books/code tables, the quantizing occurring to a word length that results in no noticeable losses in speech quality;

storing in the code books/code tables the values previously obtained by the analysis from the speech signal sampled values and used for the generation of speech signal parameters;

scaling the values of each code book/code table such that an available range of values is exploited as completely as possible, the step of scaling including the steps of:

determining a maximum of a positive value and a negative value of each code book/code table,

if the available range of values is exceeded, performing a multiplication of the values of each code book/code table by a first factor smaller than one, and

repeating the multiplication until all elements are located in the available range of values; and

causing a number of repeated multiplications to be used as a scaling factor for all code book/code table entries, wherein for a HXVC (Harmonic Vector Excitation Coding) speech coder/speech decoder, LPC coefficients, spectral envelopes of a speech signal, and unvoiced segments of the speech signal are stored in quantized form in corresponding ones of the code books/tables.

12. (Previously Presented) The method according to claim 11, wherein:

the method is performed in accordance with a method of analysis through synthesis.

13. (Previously Presented) The method according to claim 11, wherein:
the noticeable losses in speech quality are determined through a hearing test.

14. (Previously Presented) The method according to claim 11, wherein:
the first factor is 0.5.

15. (Previously Presented) The method according to claim 11, further comprising the step of:
determining word lengths of the values stored in the code books/code tables through hearing tests.

16. (Previously Presented) The method according to claim 11, further comprising the step of:
scaling the code book/code table entries to bits of a required value range.

17. (Previously Presented) The method according to claim 16, further comprising the step of:
for a finally valid quantization, performing a rounding and a subsequent truncation of decimal places.

18. (Previously Presented) The method according to claim 11, wherein:
the word length is 16 bits.

19. (Previously Presented) The method according to claim 11, further comprising the step of:
causing a processing of the code book/code table entries to occur in accordance with a digital signal processing in a whole-number format.

20. (Canceled).

21. (Previously Presented) A method for one of coding and decoding speech signal sampled values, comprising the steps of:

quantizing values previously obtained by an analysis from the speech signal sampled values and used for a generation of speech signal parameters before being stored in code books/code tables, the quantizing occurring to a word length that results in no noticeable losses in speech quality;

storing in the code books/code tables the values previously obtained by the analysis from the speech signal sampled values and used for the generation of speech signal parameters;

scaling the values of each code book/code table such that an available range of values is exploited as completely as possible, the step of scaling including the steps of:

determining a maximum of a positive value and a negative value of each code book/code table,

if the available range of values is exceeded, performing a multiplication of the values of each code book/code table by a first factor smaller than one, and

repeating the multiplication until all elements are located in the available range of values; and

causing a number of repeated multiplications to be used as a scaling factor for all code book/code table entries , wherein:

for a CELP (Code Excited Linear Prediction) speech coder/decoder, values for LSP (Line Spectral Pairs) VQ vector quantization code book/table entries, as well as those of gain VQ table entries, are stored in quantized form.

22. (Previously Presented) An apparatus corresponding to one of a coder and a decoder for processing speech signal sampled values in accordance with a method of analysis through synthesis, comprising:

an arrangement for storing in quantized form values contained in code books/code tables for a generation of speech signal parameters;

an arrangement for selecting a word length such that no noticeable losses in speech quality occur;

an arrangement for quantizing the values contained in the code books/code tables to the word length that results in no noticeable losses in speech quality;

an arrangement for scaling the values of each code book/code table such that an available range of values can be exploited as completely as possible;

an arrangement for determining a maximum of positive values and negative values of each code book/code table, and for multiplying the values of each code book/code table by a first factor less than one if the available range of values is exceeded; and

an arrangement for, if a multiplication of the values of the code books/code tables lies outside the available range of values, performing a repeated multiplication until all elements are located in the available range of values, and for providing a number of repeated multiplications as a scaling factor.

23. (Previously Presented) The apparatus according to claim 22, wherein:

the noticeable losses in speech quality are determined through a hearing test.

24. (Previously Presented) The apparatus according to claim 22, wherein:

the first factor is 0.5.

EVIDENCE APPENDIX

Appellants have not submitted any evidence pursuant to 37 CFR Sections 1.130, 1.131 or 1.132, and do not rely upon evidence entered by the Examiner.

RELATED PROCEEDINGS INDEX

There are no interferences or other appeals related to the present application.